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EXAMINER	
PATEL, HEMANT SHANTILAL	
ART UNIT	**PAPER NUMBER**
2614	

SHORTENED STATUTORY PERIOD OF RESPONSE	MAIL DATE	DELIVERY MODE
3 MONTHS	03/15/2007	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

If NO period for reply is specified above, the maximum statutory period will apply and will expire 6 MONTHS from the mailing date of this communication.

Office Action Summary	Application No.	Applicant(s)	
	10/623,613	KIM, YONG-HYUN	
	Examiner	Art Unit	
	Hemant Patel	2614	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *11 December 2006*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1,2,4,6,7,9,10 and 12-19* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1,2,4,6,7,9,10 and 12-19* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

1. The Applicant Response dated December 11, 2006 to an Office Action dated September 21, 2006 is entered. Claims 1-2, 4, 6-7, 9-10, 12-19 are pending in this application.

2. Applicant's arguments filed December 11, 2006 have been fully considered but they are not persuasive.

3. Regarding claims 1, 7, 14, 17, the Applicant argues (Remarks, pg. 14, ll. 18) "Rodgers et al. does not disclose or suggest short message service data". Examiner respectfully disagrees. The instant application specification does not disclose any specific definition, structure, format or purpose of short message service (SMS). It describes it with reference to Frequency Shift Keying (FSK) format in case of analog office line (Specification, Paragraph 33) and with reference to Pulse Code Modulation (PCM) format in case of Integrated Digital Services Network (ISDN) office line. Hence, the Office interprets it broadly as any message signal carrying any data i.e. caller identification (caller ID) signal carrying call related message data i.e. calling name, number, date and time as was well known in the art. Regarding the Applicant argument (Remarks, pg. 15, ll. 7-8) "Rodgers et al. relates to the conversion of <u>circuits</u>. This is not the conversion of <u>data</u>", Examiner would like to point out that the quoted reference (col. 17, ll. 39-64) in the previous office action clearly states "their voice paths can be connected together to complete the circuit whenever appropriate, <u>coming in via one type of interface</u>, <u>converting to standard signals</u> and <u>going out via an entirely different</u>

trunk interface being converted en route" (emphasis added) (Rogers, col. 17, ll. 50-54) and then "A special case occurs where a the call management computer 101 directly connects to and controls the organization's telephone instruments 206 with no PBX switch 204 at all" (Rogers, col. 17, ll. 55-60). Thus call management computer acts as a switch and controls connections and conversions using Digital Signal Processors (DSPs) between Central Office (CO) trunks i.e. office lines and direct connection to telephone instruments i.e. extension lines. This is similar to instant application in which incoming FSK signal SMS from analog office line is converted to or incoming PCM signal SMS is carried to PCM signal format for internal bus 170 and then converting SMS data to the format required by the terminating party i.e. single party line with Plain Old Telephone Service (POTS) telephone or ISDN line with digital terminal (Specification, Fig. 3). Regarding the Applicant argument (Remarks, pg. 15, ll. 19-20) "Rodgers et al. does not disclose or suggest switching a pulse code modulation channel from one line to a digital signal processor", Examiner respectfully disagrees. Again, the quoted reference (Rogers, col. 21, ll. 58-col. 22, ll. 4) in the previous office action clearly describes that voice paths are dynamically "switched" from one point to another and they use FMIC Flexible MVIP Interface Circuits. Rogers further describes (col. 21, ll. 48-57) that this MVIP provides 256 bidirectional voice/data channels divided into unidirectional or bidirectional "streams" of time slots, operating at 2.048 MHz. It was well known in the art that this stream format provided by MIVP was for PCM voice/data. And this switching is used when a call coming in via its CO trunk interface is connected to a DSP assigned by the call management computer (Rogers, col. 9-ll. 54-col. 10, ll. 13).

Further, the Applicant argues (Remarks, pg. 16, ll. 7-9) "Rodgers et al. does not disclose or suggest converting a pulse code modulation format short message service signal into short message service data". Examiner respectfully disagrees. Rogers clearly teaches that assigned DSPs 208 connect to respective CO trunk interfaces 203 i.e. office lines through telephony signal buses 210, and these telephony signal buses use FMIC Flexible MVIP format of PCM to convey information between external trunk interface and assigned DSP (Rogers, Fig. 2-4; col. 21, ll. 48-col. 22, ll. 4). These received information in PCM format is decoded and stored in DSP internal RAM to be accessed by call management computer 101 via PCI interface (Rogers, col. 10, ll. 3-13; col. 18, ll. 6-col. 20, ll. 50). This information is calling party identification and is received over T1 in-band signaling, which is PCM signal data as was well known in the art (Rogers, col. 24, ll. 1-7). Thus, Rogers clearly teaches of DSP receiving PCM format signal of an incoming call on an office line and converting it to a data to be stored in its memory to be accessed by call management computer processor using computer signal buses. Regarding the Applicant argument (Remarks, pg. 16, ll. 14-16) "converting a short message service data into a second pulse code modulation format short message service signal", Rogers teaches that assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory naturally needs to be converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal (Fig. 2). Regarding the Applicant arguments (Remarks, pg. 18, ll. 11-pg. 19, ll. 2) about Rand not teaching about determining

availability of a shared resource and waiting for the shared resources as claimed, Examiner would like to point out that determining an availability of a shared resource, and queuing a request for the shared resource and waiting for the shared resource to become available in a telecommunication system for efficient and economical use of shared resource was an old and very well known concept in the art. This can be for any number of functions when requestor is connected with speech path to the system and waiting for a shared resource to become available i.e. a waiting for a shared digital signal processor (digit detector) for detecting digits dialed from a touch pad of a caller's phone, waiting for a shared digital signal processor (digit outpulser) to transmit digits, waiting for another shared digital signal processor i.e. trunk circuit to make an outgoing call.

Response to Amendment

4. Applicant's arguments with respect to claims 1-2, 4, 6-7, 9-10, 12-19 have been considered but are moot in view of the new ground(s) of rejection.

Claim Rejections - 35 USC § 112

5. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

6. Claims 1-2, 4, 6 are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claim(s) contains subject matter which

was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. Independent claim 1 recites (ll. 24-26) "finishes a reception of the short message service signal if the digital signal processor resource is not available until the prescribed has elapsed". This claim recites the reception of short message service signal if the digital signal processor resource is not available but the specification discloses the reception of short message service signal _only when digital signal processor is available_ (Fig. 4A, step S102 with exit condition of "YES" only).

7. Claims 7, 9-10, 12-13 are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. Independent claim 7 recites (ll. 13-14) "finishing a reception of the short message service signal if the usable digital signal processor does not become available until the prescribed detection time has elapsed". This claim recites finishing the reception of short message service signal if the usable digital signal processor does not become available but the specification discloses the reception of short message service signal _only when digital signal processor becomes available_ (Fig. 4A, step S102 with exit condition of "YES" only).

8. Claims 14-16 are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to enable one skilled in the art to

which it pertains, or with which it is most nearly connected, to make and/or use the invention. Independent claim 14 recites (ll. 20-22) "finishes a reception of the short message service signal if the digital signal processor resource is not available until the prescribed has elapsed". This claim recites the reception of short message service signal if the digital signal processor resource is not available but the specification discloses the reception of short message service signal *only when digital signal processor is available* (Fig. 4A, step S102 with exit condition of "YES" only).

9. Claims 17-19 are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. Independent claim 17 recites (ll. 17-19) "finishes a reception of the short message service signal in the first format if the digital signal processor resource is not available until the prescribed has elapsed". This claim recites the reception of short message service signal if the digital signal processor resource is not available but the specification discloses the reception of short message service signal *only when digital signal processor is available* (Fig. 4A, step S102 with exit condition of "YES" only).

Claim Rejections - 35 USC § 103

10. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the

invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

11. Claims 1-2, 4, 6-7, 9-10, 12-19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Rogers (US Patent No. 5,946,386), and further in view of Ladd (US Patent No. 4,783,796).

Regarding claim 1, Rogers teaches of a short message service switching private branch exchange system (Fig. 1, item 101, col. 7, ll. 59-60; col. 8, ll. 6-10; PBX as integrated switch within the call management computer 101 can directly connect to extension lines), comprising:

an office line interface unit that interfaces with office lines (Fig. 2, item 203);

a voice mail interface unit including digital signal processor and a memory (Fig. 2, item 208; col. 14, ll. 25-32; col. 19, ll. 42-44), the voicemail interface unit converting a pulse code modulation format short message service signal transmitted from the office line interface unit into short message service data (col. 21, ll. 48-col. 22, ll. 4, DSP receiving MIC Flexible MVIP formatted PCM signals into data to stored in its memory; col. 18, ll. 6-col. 20, ll. 50, this data accessed by call management computer 101 through internal computer bus i.e. PCI bus; caller ID signal and data as short message service is explained above; T-1 trunks and ISDN both use PCM format for transmission was very well known in the art), and converting the short message service data into a format of a terminal that will receive the short message service data (col. 17, ll. 39-64; real-time conversion of incoming signals and data format to destination specific signals and data; and col. 17, ll. 57-60, this destination is a terminal, thus conversion will be according to format of a terminal that will receive data; col. 22, ll. 55-56, determining

called terminal; Fig. 2, assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory naturally needs to be converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal);

a control unit that switches a pulse code modulation channel of an office line to which a speech path is coupled into a pulse code modulation channel of a digital signal processor (Fig. 2, items 201, 203, 204, 210, 208; col. 21, ll. 48-col. 22, ll. 4, control unit 201 connecting incoming CO trunk interface 203 to assigned DSP 208 through circuit switch 204 and telephony bus 210; voice paths are dynamically "switched" from one point to another and they use FMIC Flexible MVIP Interface Circuits. This MVIP provides 256 bidirectional voice/data channels divided into unidirectional or bidirectional "streams" of time slots, operating at 2.048 MHz. It was well known in the art that this stream format provided by MIVP was for PCM voice/data; col. 9-ll. 54-col. 10, ll. 13, this switching is used when a call coming in via its CO trunk interface is connected to a DSP assigned by the call management computer), and determines a type of the terminal that will receive the short message service data (col. 22, ll. 55-56, determine the called party; which will be called party extension type when call management computer directly connects to extensions col. 8, ll. 6-10, ll. 20-22); and

an extension line interface unit (Fig. 2, item 206; col. 8, ll. 6-10, ll. 18-22; col. 17, ll. 57-60, call management computer directly connecting to extensions instead of trunks) that transmits a short message service signal having the format corresponding to the

terminal determined by the control unit (col. 17, ll. 39-64, real-time protocol and signal conversion; Fig. 2, assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory naturally needs to be converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal)

. wherein the digital signal processor (Fig. 2, item 208) converts the pulse code modulation format short message service signal transmitted from the office line interface unit into the short message service data by decoding (col. 19, ll. 39; col. 23, ll. 56-57, 60-61, 66-67, decoding caller ID formatted data from T-1 office lines carrying PCM data; col. 21 for T-1 trunks), converting the short message service data into a second pulse code modulation format short message service signal when the short message service data is transmitted to a single line terminal (Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory naturally needs to be converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal), and outputs the short message service data as is when the short message service data is transmitted to a digital terminal (col. 26, ll. 48-col. 27, ll. 5; notification messages sent through digital networks to workstation); and a memory that stores the short message service data (Fig. 4, item 410; col. 19, ll. 5-19; col. 20, ll. 47-50).

Rogers teaches of DSP being assigned as the incoming call (speech path) is detected (col. 9, ll. 54-62; col. 18, ll. 35-36). Thus DSP is a shared resource used for different number of incoming lines (CO trunks) and telephone lines (extension lines, col. 8, ll. 6-10, ll. 20-22). Determining an availability of and queuing and waiting for a shared resource in a telecommunication system was well known in the art and Rogers is silent about it.

However, in the same field of endeavor, Ladd teaches of queuing for and waiting for a shared resource to become available (col. 4, ll. 42-col. 5, ll. 24; offhook queuing i.e. queuing and waiting when speech path is connected).

It would have been obvious to a person of ordinary skill in the art at the time the invention was made to modify Rogers to include queuing and waiting for a shared resource as taught by Ladd in order "to provide a system for providing additional capabilities to existing PBX equipment in an office environment while not excessively increasing the cost of such upgrading of the existing equipment" (Ladd, col. 1, ll. 56-60).

Regarding claim 2, Rogers teaches of the system, wherein the office line interface unit comprises at least ***one of***:

an analog office line interface unit that couples an analog office line and converts an analog format of an short message service signal transmitted through the analog office line into a pulse code modulation format (Fig. 2, items 203, 204, 210; col. 17, ll. 39-64; col. 21, ll. 47-col. 22, ll. 4, incoming CO trunk interface 203 converting analog format on analog DID trunk to internal standardized format of MVIP); and

an integrated services digital network office line interface unit that couples an integrated services digital network office line and receives a short message service signal of a pulse code modulation format through the integrated services digital network office line (Fig. 2, items 203, 204, 210; col. 17, ll. 3-10, each office line connected with its own appropriate type of interface; col. 17, ll. 39-64; col. 21, ll. 22, ISDN PRI) and receives short message service signal (D channel and B channel transporting voice/ data over 64 kb/s PCM as very well known in the art).

Regarding claim 4, Rogers teaches of the system, wherein the extension line interface unit (Fig. 2, item 206) comprises at least ***one of***:

an single line terminal extension line interface unit (col. 8, ll. 6-10, ll. 20-22, col. 17, ll. 57-60, system connecting to telephone instruments and appearing as PBX) that couples to an single line terminal (col. 21, ll. 24-31, loopstart, groundstart for Plain Old Telephone Service (POTS) telephones) and converts the pulse code modulation format short message service signal into an analog format short message service signal by using a coder/decoder (col. 17, ll. 3-9, each appropriate interface unit 206 converting MVIP format to single line analog format); and

a digital terminal extension line interface unit that couples to a digital terminal (col. 21, ll. 46, call management computer connecting to PBX with ISDN digital line as ISDN PRI would be ISDN BRI when call management computer directly connects to telephone instruments and it appears as PBX to the telephones col. 8, ll. 6-10, ll. 20-22; col. 17, ll. 57-60; also Fig. 2, item 213, Voice over Internet Protocol (VoIP) connection).

Regarding claim 6, Rogers teaches of the system, wherein the office line interface unit comprises:

an analog office line interface unit that couples an analog office line and converts an analog format of an short message service signal transmitted through the analog office line into a pulse code modulation format (Fig. 2, items 203, 204, 210; col. 17, ll. 39-64; col. 21, ll. 47-col. 22, ll. 4, incoming CO trunk interface 203 converting analog format on analog DID trunk to internal standardized format of MVIP; col. 10, ll. 11-13, incoming path connection DSP); and

an integrated services digital network office line interface unit that couples an integrated services digital network office line and receives a short message service signal of a pulse code modulation format through the integrated services digital network office line (Fig. 2, items 203, 204, 210; col. 17, ll. 3-10, each office line connected with its own appropriate type of interface; col. 17, ll. 39-64; col. 21, ll. 22, ISDN PRI) and receives short message service signal (D channel messages or data message over B channel as was very well known in the art), wherein the voice mail interface unit comprises;

a digital signal processor (Fig. 2, item 208) that converts the pulse code modulation format short message service signal transmitted from the office line interface unit into the short message service data by decoding (col. 19, ll. 39; col. 23, ll. 56-57, 60-61, 66-67, decoding caller ID formatted data from T-1 office lines carrying PCM data; col. 21 for T-1 trunks), converting the short message service data into a second pulse code modulation format short message service signal when the short message service

data is transmitted to a single line terminal (Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory naturally needs to be converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal), and outputting the short message service data as is when the short message service data is transmitted to a digital terminal (col. 26, ll. 48-col. 27, ll. 5; notification messages sent through digital networks to workstation), and

a memory that stores the short message service data (Fig. 4, item 410; col. 19, ll. 5-19; col. 20, ll. 47-50), and wherein the extension line interface unit (Fig. 2, item 206) comprises,

an single line terminal extension line interface unit (col. 8, ll. 6-10, ll. 20-22; col. 17, ll. 57-60; system connecting to telephone instruments and appearing as PBX) that couples to the single line terminal (col. 21, ll. 24-31, loopstart, groundstart for POTS telephones) and converts the pulse code modulation format short message service signal into an analog format short message service signal by using a coder/decoder (col. 17, ll. 3-9, each appropriate interface unit 206 converting MVIP format to single line analog format), and

a digital terminal extension line interface unit that couples to the digital terminal (col. 21, ll. 46, call management computer connecting to PBX with ISDN digital line as ISDN PRI would be ISDN BRI when call management computer directly connects to

telephone instruments and it appears as PBX to the telephones col. 8, ll. 6-10, ll. 20-22; col. 17, ll. 57-60; also Fig. 2, item 213, Voice over Internet Protocol (VoIP) connection).

Regarding claim 7, Rogers teaches of a method for operating a private branch exchange system (Fig. 2, item 201), comprising:

determining whether a digital signal processor can be detected when an office line and a speech path are connected to each other (col. 9, ll. 54-62);

transmitting an short message service signal transmitted from the office line to the digital signal processor when the digital signal processor is detected (Fig. 2, item 219, col. 9, ll. 54-62, speech and control link established for transmission of data from office line 203 to DSP 208);

determining an extension line terminal that will receive the short message service signal (col. 22, ll. 55-56, determine the called party and call type); and

transmitting the short message service signal to the determined extension line terminal from the usable digital signal processor (Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory passed to the extension line interface for transmission to the extension terminal col. 8, ll. 6-10, ll. 20-22; col. 17, ll. 57-60),

wherein the digital signal processor converts the received short message service signal into short message service data (col. 24, ll. 1-7; decoding caller ID formatted data from T-1 office lines carrying PCM data or decoding data from analog DID; col. 21 for T-1 trunks), and generates an short message service message corresponding to the

extension line terminal according to a main processor (col. 17, ll. 39-64; real-time conversion of incoming signals and data format to destination specific signals and data; and col. 8, ll. 6-10, ll. 20-22, this destination is a terminal, thus conversion will be according to format of a terminal that will receive data; col. 22, ll. 55-56, the determination is based on call management computer determined called party; Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal; col. 26, ll. 48-col. 27, ll. 5; notification messages sent through digital networks to workstation).

Rogers teaches of DSP being assigned as the incoming call (speech path) is detected (col. 9, ll. 54-62; col. 18, ll. 35-36). Thus DSP is a shared resource used for different number of incoming lines (CO trunks) and telephone lines (extension lines, col. 8, ll. 6-10, ll. 20-22). Detecting an availability of and queuing and waiting for a shared resource in a telecommunication system was well known in the art and Rogers is silent about it.

However, in the same field of endeavor, Ladd teaches of queuing for and waiting for a shared resource to become available (col. 4, ll. 42-col. 5, ll. 24; offhook queuing i.e. queuing and waiting when speech path is connected).

It would have been obvious to a person of ordinary skill in the art at the time the invention was made to modify Rogers to include queuing and waiting for a shared resource as taught by Ladd in order "to provide a system for providing additional

capabilities to existing PBX equipment in an office environment while not excessively increasing the cost of such upgrading of the existing equipment" (Ladd, col. 1, ll. 56-60).

Regarding claim 9, Rogers teaches of the method, wherein when the office line is a public switched telephone network office line (Fig. 2, item 202), a switched telephone network office line interface unit (Fig. 2, item 203) converts a frequency shift keying format short message service signal to a pulse code modulation format short message service signal (col. 17, ll. 39-64; conversion to internal format of MVIP col. 21, ll. 47-57) and transmits the pulse code modulation format short message service signal to the digital signal processor (Fig. 2, items 203, 208, 204, 210; col. 21, ll. 58-col. 22, ll. 4; transmitting from CO interface 203 through circuit switch 204 and telephony signal bus 210 to DSP 208).

Regarding claim 10, Rogers teaches of the method, wherein when the office line is an integrated services digital network office line (col. 21, ll. 22, ISDN PRI), a integrated services digital network office line interface unit transmits a pulse code modulation format short message service signal to the digital signal processor (col. 17, ll. 3-9, each CO trunk including ISDN is attached through its appropriate type of interface; col. 17, ll. 39-64; trunk interface converts to internal format of MVIP col. 21, ll. 47-57; Fig. 2, items 203, 208, 204, 210; col. 21, ll. 58-col. 22, ll. 4; transmitting from CO interface 203 through circuit switch 204 and telephony signal bus 210 to DSP 208; Fig. 3, item 313; col. 20, ll. 39, voice path to DSP).

Regarding claim 12, Rogers teaches of the method, wherein the usable digital signal processor generates a pulse code modulation format short message service

signal if the extension line terminal is a single line terminal (col. 17, ll. 39-64; real-time conversion of incoming signals and data format to destination specific signals and data; and col. 8, ll. 6-10, ll. 20-22; col. 17, ll. 57-60; this destination is a terminal, thus conversion will be according to format of a terminal that will receive data; col. 22, ll. 55-56, the determination is based on call management computer determined called party; Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal), and generates the short message service data if the extension line terminal is a digital terminal (col. 26, ll. 48-col. 27, ll. 5; notification messages sent through digital networks to workstation).

Regarding claim 13, Rogers teaches of the method, wherein the pulse code modulation format short message service signal is converted into an frequency shift keying format short message service signal by a coder/decoder and the frequency shift keying format short message service signal is transmitted to the single line terminal (Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal; col. 15, ll. 26-34; col. 22, ll. 55-56; col. 8, ll. 6-10, ll. 20-22; col. 16, ll. 4-6; col. 17, ll. 57-60; telephones connected to call management computer).

Regarding claim 14, Rogers teaches of a method for switching short message service of a private branch exchange system, comprising:

switching a pulse code modulation channel of an office line interface unit to a pulse code modulation channel of a digital signal processor if a speech path is connected to the office line interface unit (col. 9, ll. 42-62; col. 21, ll. 48-col. 2, ll. 4; connecting DSP to trunk interface through circuit switch and telephony signal bus as needed; col. 10, ll. 1-13, incoming path connection to DSP; col. 20, ll. 39);

transmitting an short message service signal to the digital signal processor from the office line interface unit through the pulse code modulation channel (Fig. 2, items 202, 203, 204, 210, 208, any data on incoming trunk 202 path transmitted to DSP 208 through trunk interface 203 and circuit switch 204; col. 17, ll. 39-64; conversion to internal format of MVIP col. 21, ll. 47-57);

decoding the short message service signal transmitted to the digital signal processor (col. 24, ll. 1-7; decoding caller ID formatted data from T-1 office lines carrying PCM data or decoding data from analog DID; col. 21 for T-1 trunks);

switching the pulse code modulation channel of the digital signal processor to a pulse code modulation channel of an single line terminal extension line interface unit if an extension line terminal that will receive the short message service signal is an single line terminal (Fig. 2, items 208, 204, 210, 206; col. 8, ll. 6-10, ll. 20-22; col. 16, ll. 4-6; col. 17, ll. 57-60; connecting to telephones; col. 22, ll. 55-56, called party determination by call management computer); and

switching an short message service data channel of the digital signal processor to an short message service data channel of a digital terminal extension line interface unit if the extension line terminal that will receive the short message service signal is a digital terminal (col. 26, ll. 48-col. 27. ll. 5).

Rogers teaches of DSP being assigned as the incoming call (speech path) is detected (col. 9, ll. 54-62; col. 18, ll. 35-36). Thus DSP is a shared resource used for different number of incoming lines (CO trunks) and telephone lines (extension lines, col. 8, ll. 6-10, ll. 20-22). Determining an availability of and queuing and waiting for a shared resource in a telecommunication system was well known in the art and Rogers is silent about it.

However, in the same field of endeavor, Ladd teaches of queuing for and waiting for a shared resource to become available (col. 4, ll. 42-col. 5, ll. 24; offhook queuing i.e. queuing and waiting when speech path is connected).

It would have been obvious to a person of ordinary skill in the art at the time the invention was made to modify Rogers to include queuing and waiting for a shared resource as taught by Ladd in order "to provide a system for providing additional capabilities to existing PBX equipment in an office environment while not excessively increasing the cost of such upgrading of the existing equipment" (Ladd, col. 1, ll. 56-60).

Regarding claim 15, refer to rejection for claim 14 and claim 9.

Regarding claim 16, Rogers teaches of telephones (extension lines) attaching to management computer with their appropriate interface type cards (col. 17, 3-9, interface types; col. 8, ll. 6-10, ll. 20-22 connecting telephones). The extension interface

cards were known in the art to generate frequency shift keying signals for transmission of a data message to destination single line POTS telephone terminal. This is part of Custom Local Area Signaling System (CLASS) features as per industry standard TR-NWT-000031 that uses signal transmission performed according to accompanying industry standard TR-NWT-000030.

Regarding claim 17, Rogers teaches of a private branch exchange system, comprising:

a single digital signal processor that receives a short message service signal in a first format and converts the short message service signal into a second format short message service signal (col. 17, ll. 39-64, DSP converts received data from internal format of MVIP to format of receiving terminal i.e. digital terminal col. 26, ll. 65-col. 27, ll. 5; col. 24, ll. 1-7; decoding caller ID formatted data from T-1 office lines carrying PCM data or decoding data from analog DID; col. 21 for T-1 trunks; col. 8, ll. 6-10, ll. 20-22; col. 15, ll. 26-34; col. 17, ll. 57-60; the destination is a terminal, thus conversion will be according to format of a terminal that will receive data; Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal; col. 26, ll. 48-col. 27, ll. 5; notification messages sent through digital networks to workstation); and

a controller (Fig. 2, item 201) that controls the digital signal processor (col. 9, ll. 54-62; col. 18, ll. 35-36; assigns DSP as needed thus it controls them) and determines the second format (col. 22, ll. 55-56; determines called party and thus required format),

wherein the digital signal processor converts the first format short message service signal into the second format short message service signal by converting the first format short message service signal into short message service data (col. 24, ll. 1-7; decoding caller ID formatted data from T-1 office lines carrying PCM data or decoding data from analog DID; col. 21 for T-1 trunks and stored in DSP memory) and converting the short message service data to the second format short message service signal (Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory converted to PCM format signal of telephony signal busses to be passed to the extension line interface for transmission to the extension terminal; col. 26, ll. 48-col. 27, ll. 5; notification messages sent through digital networks to workstation), the converting the first format short message service signal into the short message service data comprising decoding (col. 24, ll. 1-7; decoding caller ID formatted data from T-1 office lines carrying PCM data or decoding data from analog DID; col. 21 for T-1 trunks and stored in DSP memory), converting the short message service data into the second format short message service signal when the short message service data is transmitted to a single line terminal (Fig. 2; assigned DSP 208 connects to telephones i.e. extension line through telephone signal busses 210 and interfaces 206, and the decoded caller ID data stored in DSP memory converted to PCM format signal of

telephony signal busses to be passed to the extension line interface for transmission to the extension terminal; The extension interface cards were known in the art to generate frequency shift keying signals for transmission of a data message to destination single line POTS telephone terminal. This is part of Custom Local Area Signaling System (CLASS) features as per industry standard TR-NWT-000031 that uses signal transmission performed according to accompanying industry standard TR-NWT-000030), and outputting the short message service data as is when the short message service data is transmitted to a digital terminal (col. 26, ll. 48-col. 27, ll. 5).

Rogers teaches of DSP being assigned as the incoming call (speech path) is detected (col. 9, ll. 54-62; col. 18, ll. 35-36). Thus DSP is a shared resource used for different number of incoming lines (CO trunks) and telephone lines (extension lines, col. 8, ll. 6-10, ll. 20-22). Determining an availability of and queuing and waiting for a shared resource in a telecommunication system was well known in the art and Rogers is silent about it.

However, in the same field of endeavor, Ladd teaches of queuing for and waiting for a shared resource to become available (col. 4, ll. 42-col. 5, ll. 24; offhook queuing i.e. queuing and waiting when speech path is connected).

It would have been obvious to a person of ordinary skill in the art at the time the invention was made to modify Rogers to include queuing and waiting for a shared resource as taught by Ladd in order "to provide a system for providing additional capabilities to existing PBX equipment in an office environment while not excessively increasing the cost of such upgrading of the existing equipment" (Ladd, col. 1, ll. 56-60).

Regarding claim 18, Rogers teaches universal conversion signals and voice from any trunk/interface to any other trunk/interface (col. 17, ll. 39-60).

Regarding claim 19, Rogers teaches of DSP with a voice mail interface (col. 14, ll. 24-32; col. 19, ll. 42-43).

Information Required - 37 CFR 1.105

12. Examiner believes that a complete and detail search has been performed but does not discover any references that relate to switching system such as PBX providing short message service as discussed on pages 1-5 of the specification as admitted prior art.

13. Applicant and the assignee of this application are required under 37 CFR 1.105 to provide the following information that the examiner has determined is reasonably necessary to the examination of this application.

14. The information is required to identify products or services embodying the disclosed subject matter stated on pages 1-5 of the specification, particularly Short Message Service Signals and Short Message Service Data definition, description, format and encoding in relation to Private Branch Exchange system or Public Switched Telephone Network, or identify the properties of similar products and services found in the prior art.

15. In response to this requirement, please provide the title, citation and copy of each publication that is a source used for the description of the prior art in the disclosure.

16. The fee and certification requirements of 37 CFR 1.97 are waived for those documents submitted in reply to this requirement. This waiver extends only to those documents within the scope of this requirement under 37 CFR 1.105 that are included in the applicant's first complete communication responding to this requirement. Any supplemental replies subsequent to the first communication responding to this requirement and any information disclosures beyond the scope of this requirement under 37 CFR 1.105 are subject to the fee and certification requirements of 37 CFR 1.97.

17. The applicant is reminded that the reply to this requirement must be made with candor and good faith under 37 CFR 1.56. Where the applicant does not have or cannot readily obtain an item of required information, a statement that the item is unknown or cannot be readily obtained may be accepted as a complete reply to the requirement for that item.

18. This requirement is an attachment of the enclosed Office action. A complete reply to the enclosed Office action must include a complete reply to this requirement. The time period for reply to this requirement coincides with the time period for reply to the enclosed Office action.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Hemant Patel whose telephone number is 571-272-8620. The examiner can normally be reached on 8:00 AM - 5:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Fan Tsang can be reached on 571-272-7547. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

Hemant Patel
Examiner
Art Unit 2614

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